
# The beginners' guide to EFFECTS

In the first of a series of beginner tutorials introducing five fundamental computer music concepts, we show you how to improve your sounds with effects plug-ins

**cm DVD**
**ON THE DVD**
The audio files referred to in the walkthroughs can be found in the **Tutorial Files** folder

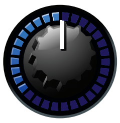
**> Just what makes a production a production? What is it that gives a professionally recorded and mixed song that special, elusive power and punch, depth and detail? You've got the killer beats and wicked loops. You've got all of the instruments: samplers, synths, drum machines abound. You've got the songwriting and performance chops. Yet there's something missing. The tracks lack a sense of space. They lack interest, falling flat the moment the sound drops out of the speakers. Why? What do the pros know that you don't?**

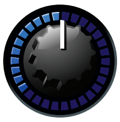
In a word: effects. Effects can make (or break) a track. They can be the perfect sweetener, adding just the right amount of gloss or grit to bring your songs to life. They're a crucial ingredient in virtually any pop, rock or dance production – even classical music has some reverb thrown on, albeit of the natural variety.

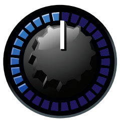
For decades now, hardware studio effects units have emulated acoustic and mechanical processes, and these sorts of effects remain among the most popular: the reverberation of a natural acoustic space, for example, the whooshing jet noise effect of tape flanging, or the nasty distortion of an overdriven amplifier. Today, software plug-ins bring these real-world devices into the virtual domain, with no compromise whatsoever in terms of features and sound quality. There's a vast wealth of effects plug-ins available, both free and for paid-for. Your DAW almost certainly includes a suite of them, and there's a fine selection in the **cm** Studio on the *Computer Music* DVD each and every month.

So how can you get the most from them? How are you to know what effects should be used and when? Over the next ten pages, we're going to guide you around the most common and frequently applicable types of effects and give you step-by-step instruction on how they might be used. We'll also show you some more out-there plug-ins and discuss their role in modern music production. As you explore each of these effects, you'll begin to realise that you've known about them all along, even if you didn't have a name to describe them, and by the time you've worked through this tutorial, you'll have the knowledge and tools to make your projects more engaging and professional.

# Delay and reverb







We begin with two of the most common (and essential) effects. The first of these, delay, is closely related to the second, reverb, although far simpler in design. A delay plug-in does exactly what the name implies: it delays the audio signal by a specified amount and plays it back at a later point in time. A delay can be used to adjust timing errors in a signal or performance, but it's more commonly used to create an echo effect. You've heard it countless times: the subtle slapback echo on a surf rock guitar, or the spacey tail of a trance synth lead.

The first delay units were tape-based devices. In fact, some delay plug-ins continue to present themselves in faux tape-machine finery, complete with movable tape heads to adjust the time between each repeat of the original signal.

Just about every delay plug-in enables you to control the delay time (most can be synchronised to the tempo of the DAW hosting the plug-in), along with a mix control to determine how much of the original signal is audible in relation to the processed signal (called the 'dry' and 'wet' signals – remember that, because it comes up again and again).

There will also very likely be a feedback control, which enables you to route the processed signal back through the delay again for thicker, more intense effects. Be careful, though: too much feedback can lead to uncontrollably loud signals that can damage your ears and your speakers.


**Throw your echoes around and around and infuse them with some grit with the superb (and free!) TAL-Dub-2**

Finally, your delay plug-in may have a built-in filter, which alters the frequency content of the echoes to better simulate the sound of hardware echo boxes, or can be used as a sound-shaping tool.

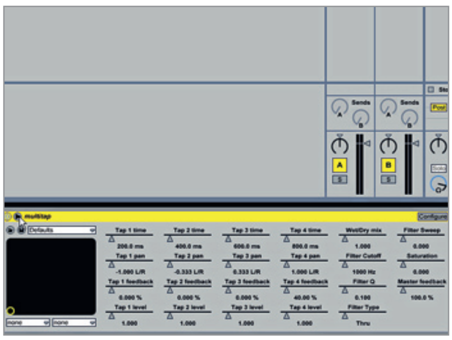
## > Step by step Tape delay


1 > Delay is used for everything from simulated double tracking to deep, cosmic echoes. There's even an entire subculture of musicians using looping delays as the basis of their sound. Fire up your DAW, and call up an electric piano instrument, such as this one in Logic.

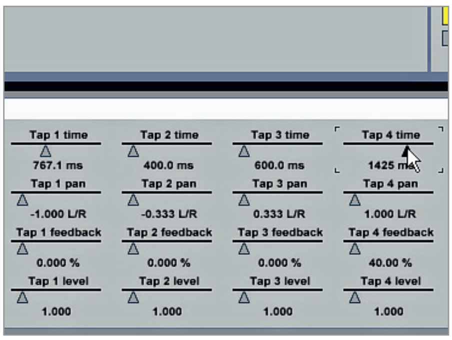

2 > Make sure that there's no built-in echo (or any other effects, for that matter) active in the instrument. There are plenty of delay plug-ins that simulate tape echo. We're going to use Logic's Tape Delay, but you shouldn't have any trouble finding something like it in any other DAW – or grab TAL-Dub-2 from **kunz.corrupt.ch**.


3 > Play a staccato chord and you'll hear the echoes fading away in sync with your DAW's tempo. Now reduce the **High Cut** to around **360Hz** to emulate the tone degradation of tape-based echo. To simulate tape-like motor imperfections, increase the **LFO Depth** a little, which will subtly change the pitch of the echoes.

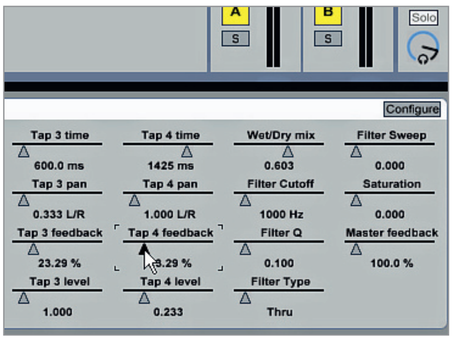
## > Step by step Multi-tap delay



1 > Multi-tap delays enable you to position multiple delayed signals, creating complex rhythmic echoes. Expert Sleepers offer a magnificent multi-tap unit for free (**www.expert-sleepers.co.uk**). Throw it onto a piano sound and play a staccato chord – the effect is very different to that of a tape delay, as you can hear.




2 > Let's adjust the delay time of the 'taps' to create a funky, interesting rhythm. Push the **Tap 1 Time** slider up past **760ms**, and the **Tap 4 time** to **1425ms**. Try another chord and you should hear some staggered, polyrhythmic echoes as the plug-in triggers repeats at various points after the original signal.

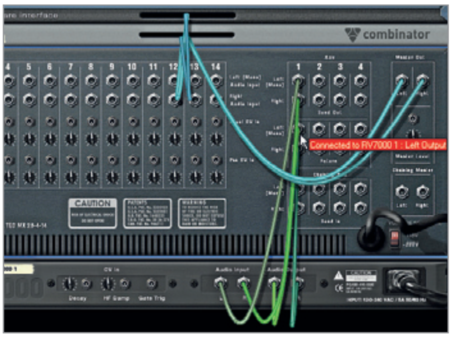



3 > Adjust the **Wet/Dry Mix** to **0.603** to hear more of the original signal. A multi-tap plug-in like this one will enable you to take control of each individual tap – you'll notice that they're already panned. Play around with the **Feedback** and **Level**s for each tap, but be especially careful with the former.

## > Step by step

### Reverb basics

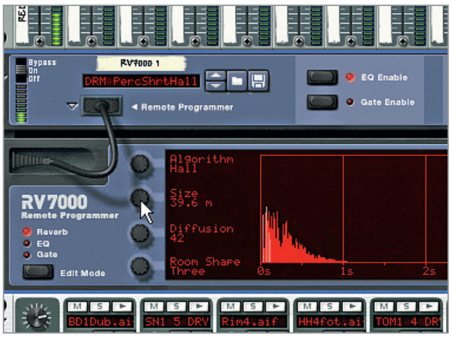

**1** > Your DAW will undoubtedly have a reverb included. We're going to use Reason's RV7000 for this walkthrough, though you can follow along with any reverb in any DAW, since the parameters we're using are common to all of them. Fire up a new project and **Create** a Mixer 14:2 and a RV7000 reverb device.



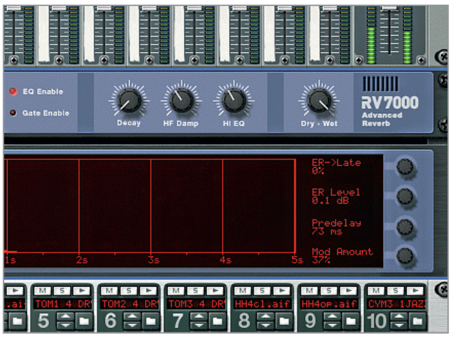

**2** > Let's take a moment to look behind the Reason rack – hit the **Tab** key. Notice that our reverb unit is being routed to the Aux Sends and Returns. If you're not using Reason, you may have to create an aux bus and stick your reverb on that. Every host takes a different approach, so consult your host's manual, if you don't know how to do this.


**3** > Now, let's add a Redrum drum machine. Load up the **Dublab TightKit1** kit from the **Factory Sound Bank**. If you're not using Reason, load any drum machine and select a dry kit (ie, one with clean, unprocessed sounds – some samples come with reverb recorded into them, which we want to avoid). Now program a basic kick/snare pattern.


**4** > Let's choose a patch for the RV7000. Try **PercShrtHall** from the **DRM** category. Play your drum beat. It's still dry and thus sounds dull and lifeless. Find the Aux send knobs at the top of the mixer. Our reverb is on Auxiliary 1. Push the **Channel 1 Aux 1 Send** knob past the half-way mark – we've gone up to **85**.




**5** > That's much better! Our drums now have the illusion of being in an acoustic space. The RV7000 is an algorithmic reverb and creates this virtual space using a series of echoes. We can change the Algorithm and adjust the room Size – let's do the latter. Click the **Remote Programmer** arrow to put it into Edit mode, and give the **Size** dial a nudge. Your acoustic space just got bigger!




**6** > Finally, have a play with some of the other parameters. **Predelay** is very important in creating a good ambient sound – increasing it enables the original signal to retain some clarity – while adjusting the **Diffusion** smooths things out a bit. Experiment further as you like.


## Sends and inserts

**You may well have noticed that your mixer enables you to apply effects as 'inserts' and 'sends' (aka, 'auxiliaries'). So what's the difference? Well, with an insert, the effect (or chain of effects) is only applied to the channel into which it's loaded, and the entire signal is processed – although if the plug-in has a wet/dry control, you can set the audible balance between the original signal and the effected one.**

**A send effect, on the other hand, is placed on an auxiliary bus that's available to every channel in the mixer via each channel's individual 'send' control, which enables you to send as much or as little as you like of every sound in your track to that effect. Thus, you can use, say, a single reverb to give the whole track a sense of space, with some sounds getting just a little bit of it and others getting more. This is especially useful if your reverb plug-in is CPU-intensive, since you're only using one of them to effectively do the work of many.**

**As mentioned before, the wet/dry mix enables you to set the balance of an insert effect. When using an effect on an auxiliary bus, the send controls on the mixer are used to determine the ratio of processed/unprocessed signal.**

**Effects that are intended to process 100% of a single signal are typically used as inserts (distortion, EQ, compression), while others are better suited for sends (reverb, delay), but there are, of course, no hard and fast rules.**

## Convolution and algorithmic reverb

**Not so long ago, algorithmic reverb was the only kind of digital reverb available. It simulates complex acoustic spaces by piling on lots and lots of delays with subtle variations to their feedback and frequency. Depending on the algorithm used, the effect can be quite convincing, and algorithmic units from companies like Lexicon and TC Electronic set the standard for studio reverb for many years.**

**Convolution reverbs, however, use sampled 'impulse responses' of real spaces to create a complex mathematical model of how your signal might sound in such a space. The results are nothing short of remarkable in terms of realism and quality, and they're very different to what you get from an algorithmic reverb. Nevertheless, both types have a place in modern production.**

# Creative vs corrective effects processing

**The subject of EQ provides the perfect opportunity to discuss the dual nature of certain effects processors, including compressors, limiters, exciters and even pitch correctors. All of these share a certain duality: they were originally designed to correct problematic recordings or performances, but turned out to also offer plenty of creative potential.**

**EQ, for example, was created to give the engineer the ability to attenuate (reduce) offending, overpowering frequencies and boost those that are weaker than they ought to be. Similarly, compressors and limiters are intended for correcting wayward fluctuations in volume, while the intended role of pitch correction is self-evident. Yet each and every one of these effects has become a creative tool. Engineers and producers have been quick to seize upon processor design quirks and subvert them to more novel usage.**

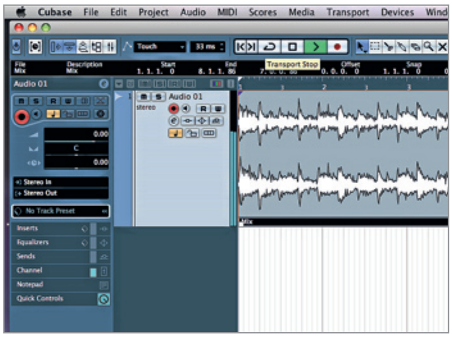
**The Auto-Tune effect is a prime example, as is the ever-popular filtered drum loop, wherein a filter (which is, fundamentally, an EQ) is used to remove much of the frequency content, leaving the loop with a highly stylised sonic signature. Then there's the familiar 'telephone effect' that's frequently applied to vocals or guitars. This effect is created with an EQ or filter and is anything but corrective.**

**Compressors and gates are also often used for purely creative purposes. You can't miss the percolating pulse of the 'trance gate', so prevalent in modern dance music. This effect is created by sidechaining one sound into an effect to control how that effect is applied to another sound. Indeed, sidechaining itself was originally designed as a corrective tool. It enabled the 'ducking' (lowering in volume) of one signal when another one's volume reached a certain threshold. It was meant to be used to reduce the volume of program material when a voice entered the scene – a radio announcer talking over music, for example.**

**The corrective roles of these effects are still essential. Use them sparingly, but don't overlook the possibility of applying them in more creative ways. Explore them by tweaking and automating every parameter available. You never know, you might stumble upon that Next Big Thing in music production.**

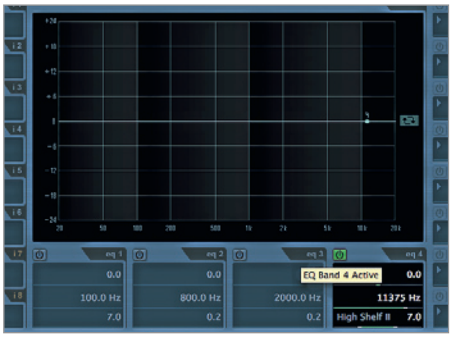
## > Step by step

### Using parametric EQ

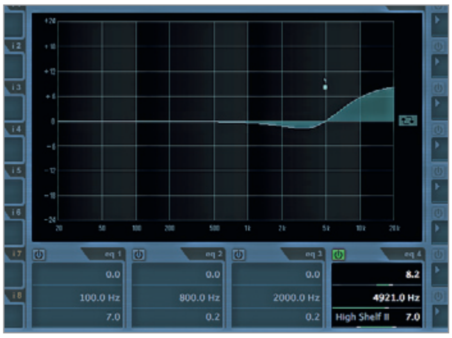


**1** > You'll no doubt be familiar with EQ already, since it's been a standard feature of 'domestic' music playback systems for years. However, that sort of EQ is usually of the 'graphic' variety, with fixed frequency bands, or even just bass and treble controls. Engineers need something a more flexibility in the form of parametric EQ. Import **Mix.wav** from your **cm** DVD into your DAW of choice.

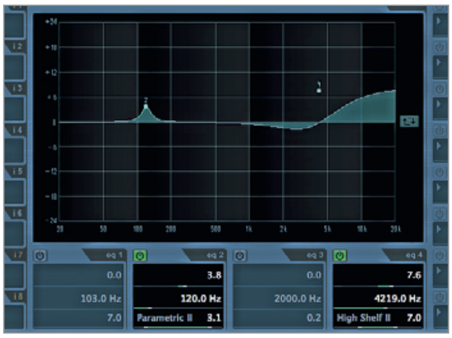
**2** > Take a listen to the clip. It sounds pretty good, but there's something missing from the higher frequencies, and there's a bit of muddiness in the low-mids. Let's try finessing it with a little EQ. This sort of job requires the precision of a parametric equaliser...



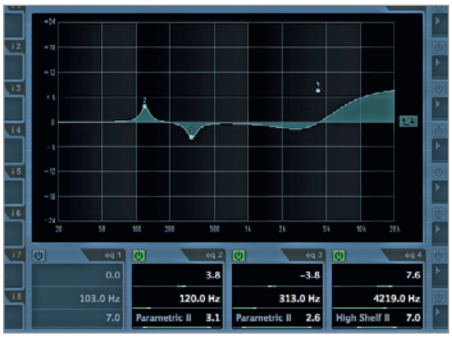
**3** > If you're using Cubase, as we are, click the channel **E**dit button to bring up all of the channel's parameters. You could use the Inspector, but our method gives us a clearer picture of our equaliser. You can, of course, use any parametric EQ you like, but we're going with the one in Cubase. Activate EQ Band 4.



**4** > Band 4 is currently set to act as a high shelf, meaning that everything above the selected frequency will be affected. This suits our purposes, since we want to add a little high-end 'air' to the mix. We boost the area around **4900Hz** by a whopping **8.2dB**. This is ridiculously heavy-handed, but we want you to hear the effect clearly.

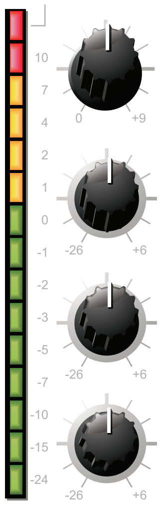


**5** > Let's pull the range down to **4200Hz** at a level of **7.6dB**. Now, let's give a little bump to the bass. We'll reserve Band 1 for a low shelf in case we need to cut out some rumble later on, so let's use Band 2. Give it a slight nudge at **120Hz – 3-4dB** should suffice. Set the **Bandwidth** (aka, **Q**) slider at the bottom to **3.1**. This controls how much the frequencies immediately surrounding the target area are affected.



**6** > Now our low end is a little beefier, and our high end a little airier. There's still a lack of clarity in the lower-mids, though, so activate Band 3, and reduce the **200-500Hz** range by about **3dB**. As you can hear, this removes some of the murkiness. Our mix now makes a lot more sonic sense than it did previously.

# Compressors, gates and limiters



This is the class of corrective devices known as 'dynamics processors', designed to control the volume of your signals. For example, have you ever found that the volume of your drums or vocals is all over the place? Enter the compressor, which gives a more uniform level throughout by compressing the volume range of the signal, making the quiet bits louder and the loud bits quieter. You're (usually) given control over the compression ratio (strength), threshold (volume level) at which compression should begin to be applied, and make-up gain, which brings the level up after the compressor has done its work, since the process invariably leads to a certain loss in overall volume as well as a reduction in distance between the loudest and quietest bits.

A limiter is another, simpler type of compressor. It simply creates a 'ceiling' beyond which the signal will always be reduced in volume to a predetermined level. Multiband compressors, meanwhile, apply different amounts of compression or limiting to different frequency ranges, and are used in the mastering process to make the mix sound louder than it actually is.

A gate is a little different, but shares many of the same parameters as a compressor. It simply shuts the output off entirely when the signal falls below a certain threshold. This is commonly used to allow the desired parts of a signal to pass through, while blocking any lower-level noise in between them.


**Impart some of that always-desirable retro character to your recordings with PSP Audioware's plug-in classic, VintageWarmer**

## > Step by step

### Applying compression


**1** > Compression is a fairly simple process, but it often confuses many novice engineers. Let's take a look at the basics. Launch your DAW and import the audio file **Piano.wav** from your **cm** DVD. Have a listen to the sample. Obviously, the levels are all over the place. We can use CompressiveCM (in the **CM Studio** folder on the DVD) to give them some consistency, though.


**2** > CompressiveCM has all the functions you'd expect in a dynamics plug-in (including sidechaining, which we'll get on to shortly). We only need the top section, however, for basic compression. Play your sound, then lower the compressor's **Threshold** by a considerable margin and listen to the dynamics. We've set ours to **-30dB**. Anything entering the plug-in over that level will now be compressed.


**3** > Can't hear any difference? That's because the **Ratio** is set to **1:1**, meaning that for every 1dB the signal goes over the threshold, it's raised by 1dB (ie, it stays the same). Raise it to **9.35:1**. Now the signal exits the compressor only 1dB louder for every 9.35dB it enters it over the threshold level. Crank up the **Master** volume to make up for the reduction. Now we have a more uniform level throughout.


**4** > You might have noticed, however, that pushing the **Ratio** up so far has brought some of the harshness of the piano's attack to the fore. We've really only pushed it this high to be sure that you could hear the effect clearly. Scale it back to **3.66:1**. The **Attack** and **Release** controls govern how long the compression will take to kick in whenever the signal exceeds the threshold.


**5** > Compressors often have an adjustable 'knee'. This determines whether the compressor's response is sharp or gradual – the softer the knee, the more gradual the rise to full effect will be. Turn the **Knee** knob up to around **66%**. Here, the difference is subtle, but it can be quite dramatic, depending on the source material. Finally, set the **Valve** to **Dirty** mode for a vintage compression effect.

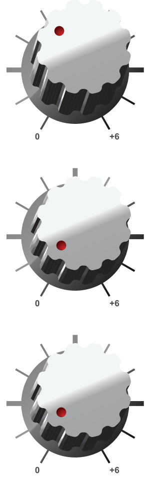
POWER TIP

## >Sidechaining

An effect with a sidechain input enables you to use the volume of one signal to control the processing of another. For example, you could run a drum loop into the sidechain input on a gate plug-in processing a synthesiser sound. When the drum loop's volume exceeds the set threshold, the gate will open, allowing the synth signal to pass through. This can be used creatively, as described here, or in a corrective way. You could, for example, use sidechaining to lock a bass sound to a kick drum track.

Lots of different effects have sidechain inputs – compressors, limiters, filters, etc. It's no wonder that sidechaining is used by so many producers.

# Chorus, phasers and flangers



This class of effects all draw upon the same processes to produce distinctly different results. However, their origins and intent are exclusive to each.

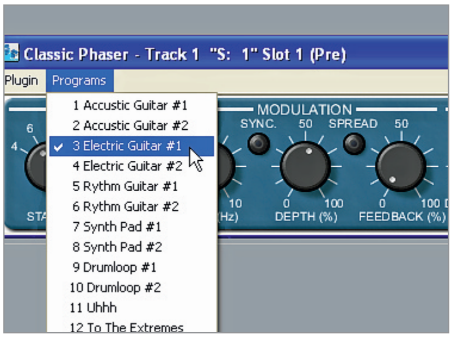
The phaser is based on the idea that each waveform has a distinctive cycle, made up of positive and negative phases (usually identical, with one of them 'flipped'). A signal's phase is an important consideration in getting a good recording, and engineers are always battling with phasing issues introduced by microphones. Conversely, we can play with a signal's phase to produce interesting results. Here's the thing: when you play an exact copy of a signal alongside itself and reverse the phase of one of them, the signals will cancel each other out, resulting in complete silence. If the copied signal is then sent through an all-pass filter (**en.wikipedia.org/wiki/All-pass_filter**), with an LFO to modulate (change over time) the frequency, you'll get that characteristic sweeping effect.


Add some psychedelic flavour with phasing plug-ins like D16's Fazortan

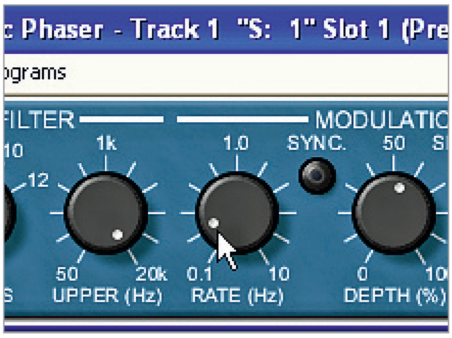
The sound of the flanger is similar to that of the phaser, except that a very short delay is placed in the signal path. This provides a reasonable simulation of tape flanging, an effect created by playing two tapes with identical material recorded onto them at the same time, and manually altering the speed and frequency content of one deck to get the classic 'jet plane' sound so adored by psychedelic musicians.

Finally, chorus. This one uses a similar method to flanging to emulate the sound of multiple sources, each slightly different in pitch from the others. The reality is a shimmering kind of sound that works wonders on guitar, electric piano and the relatively thin timbres of simple analogue waveforms.

## > Step by step Phasing a guitar




**1** > Phasing is a simple effect that can dramatically improve a sound. Import the file **12String.wav** into your DAW from the DVD and insert a phaser plug-in. Kjaerhus' (**www.kjaerhusaudio.com**) Classic Phaser is a fine contender. We've chosen an electric guitar preset.

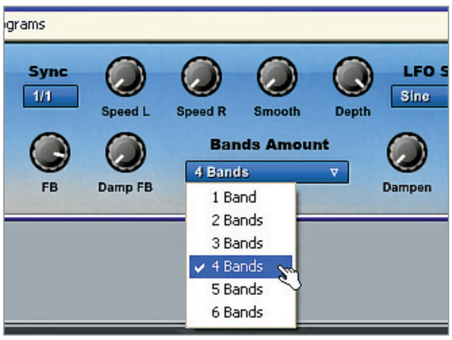



**2** > Phasers often have multiple 'stages'. This refers to the number of band-pass filters onboard and the number of 'notches' they make in the sound. More stages means a more complex sound. Switch to **12 Stages**. Also, give the **Rate** knob a downward twist for a slow, deep sweeping effect.


**3** > That's pretty subtle, so increase the intensity by cranking the **Depth** knob up to 3 o'clock. For some extra drama, toggle in the **Spread** function, and for a classic whooshing sound, increase the **Feedback** amount – try boosting it past the halfway mark, then tune up, turn on and freak out!

## > Step by step Flanging fundamentals


**1** > Let's move on to the phaser-related effect called flanging. Use the same clip as above, and preferably the same host, only this time, replace the phaser effect with a flanger. We're using the free Flang-3R (**www.atomsplitteraudio.com**). You'll hear a deeper, richer sound with the flanger than you did with the phaser.

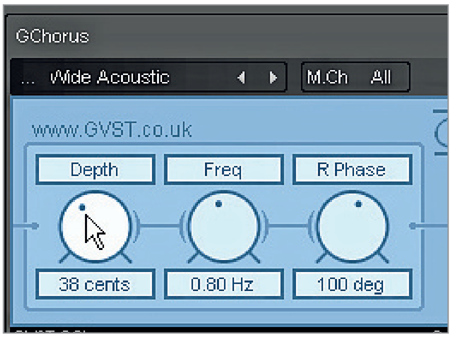



**2** > Play the default setting. This effect is often mistakenly referred to as phasing when it is, in fact, flanging. The difference is clear, isn't it? Adjust the **Shift** knob for a more extreme effect. Still not enough? You want the legendary 'jet plane' sound, right? Try increasing the **Bands Amount** to **4** for a deeper effect.

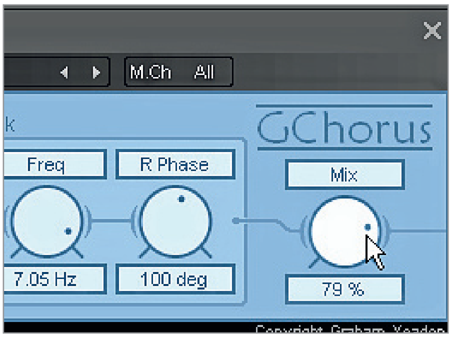

**3** > Currently, the speed of the LFO is locked to the host's tempo. That's fine for some applications, but we'd rather take the reigns ourselves. Click the **Mode** menu and select **Speed**. Now, try adjusting the **Speed L** and **R** knobs to change the rate of flanging independently for both sides of the stereo soundstage.

## > Step by step Chorus


1 > Chorus is closely related to phasing and flanging, so we're going to use the same audio file to see what it can do. Load the sample into your DAW of choice and stick a chorus plug-in on an insert. We're using GVST's GChorus (**www.gvst.co.uk**).




2 > Listen to the default patch. It's thicker and more complex than the phaser, but not as extreme as the flanger. It sounds a little like a double-tracked recording (that's the idea), but misses the mark in some interesting ways. We can get dangerously close to Pink Floyd by increasing the **Depth**.




3 > Boost the **Frequency** up to **7Hz** or so, and turn the **Mix** up to **79%**. It's starting to sound like a rotary speaker! In fact, you can imitate that particular retro favourite with a pair of chorus effects, each set to different speeds and frequencies. Experiment with the settings to get a feel for the possibilities.

## > Step by step Distortion in brief


1 > While we're on the subject of guitars, we really ought to take a look at some of the things you can do with distortion. Remove your chorus plug-in and replace it with a distortion effect, such as the fantastic CMFuzz, in the **CM Studio** folder on the DVD. This is a great processor with a trio of effects onboard.


2 > Flip through the presets. Most of them are too heavy for our sombre guitar recording. However, the **Tubular** patch works pretty well. It combines distortion and compression. Set the **Tube** amount to about halfway. Do likewise with the Compressor section's **Amount** knob. This effect sounds great on guitar.


3 > There are many amp sims out there. We've got a great collection of famous guitar gear emulations called Guitar Suite CM. Drag the JCM900 into an insert after our fuzz box. This is a copy of a very popular Marshall guitar amp. Tweak the sliders to hear what it can do, and strike a rock god pose. Nice.

## Delay-based processing and more

**There are only a handful of different types of effect available, and most plug-ins are built on one or another of them. Among the most prominent types is the delay-based (aka time-based) effect. As we mentioned before, chorus, flanging, phasing and reverb plug-ins employ delay to process sounds, and so we tend to group these together under that heading.**

**We probably needn't tell you what the term 'pitch-based' describes, and that particular type is often combined with time-based processing in plug-in design. In fact, the two used to be inextricably linked. However, recent developments in DSP (Digital Signal Processing) technology have created a class of pitch effects that are now removed from the time domain. Until not that long ago, any shift in pitch resulted in a similar shift in time. In other words, if you wanted to, say, reduce the pitch of a sound by an octave, that sound would have to be stretched to twice its original length to make it happen. Now we can make use of various time and pitch altering processes that change one without affecting the other. Granular effects fall into this category, as do 'elastic audio' programs like Acid and Live.**

**Distortion effects are entirely different, relying on various different processes to distort the original signal. Some emulate physical phenomena, such as overdriven amplifiers, while others exert influence over the mathematics of digital audio in order to degrade it. Fuzzboxes, plus tube and tape saturation belong to the former bunch, while bit-crushers and aliasing plug-ins fall into the latter group. These all sound very different to one another. It's worth noting that distortion is sometimes added to non-distortion effects – many compressors, EQs, delays, etc, feature 'tube' or 'saturation' settings, designed to add a bit of vintage-style grit to their sound.**


**Sanford Phaser is one big, bold plug-in. It's also built around delay, though that might not at first be obvious**


An effect like this can add interest, but show a little restraint, yeah?

# Don't go overboard

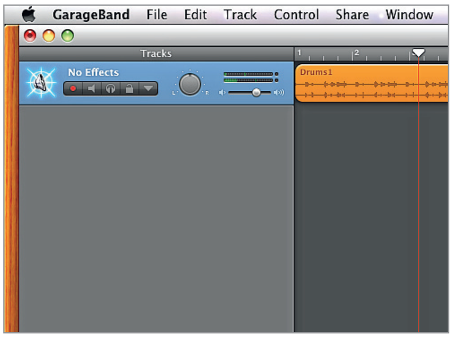
**From radio to the internet to the television, we're inundated with tunes these days, from bedroom demos to professionally mixed masterpieces – and one of the main things that differentiates the two is the usage of effects. A professional engineer knows how to use effects with discretion and taste. He or she knows that subtlety is the key to a successful mix. Reverb is used to suggest a space, delay to add a bit of drama here and there. Dramatic special effects should be used sparingly. When an effect is overused or abused, it loses its potency. We don't have to tell you that. It's been a long time since any of us thought an exaggerated, Auto-Tuned vocal was a fresh idea. Effects are the icing on the cake. The right amount of them, perfectly chosen, can accentuate and draw out the flavour of your track. However, if you add too many effects, or choose the wrong types, you'll end up annoying your audience and dissuading them from further listening.**

**It can be tempting to pile on as many effects as you have, turn all the dials up to 11 and let them run for the entirety of a mix, but such gimmickry will grow tiresome pretty quickly. Remember that, just because it sounds cool, doesn't mean you need to use it all the time.**

**Exercise restraint and know your threshold for ear fatigue. And take frequent breaks when mixing, so that you don't lose focus and perception. Reverb is a particularly insidious effect, for example, since it simulates a natural phenomenon. After an extended mixing session, your ears become accustomed to it and you don't notice it, so you reach for the aux send and add a little more. Pretty soon, the whole track is swimming in the stuff. EQs and filters can have similar issues, so relax and go easy. When you think you have enough reverb, dial it down by 10%. You'll be glad you did.**

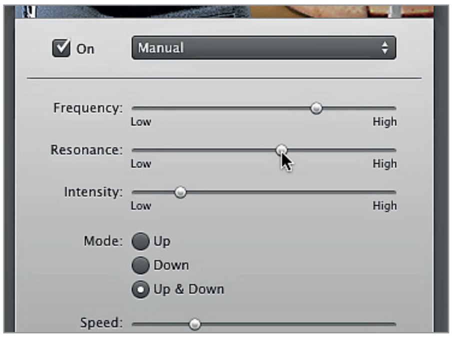
## > Step by step

### Filters – EQ but more fun

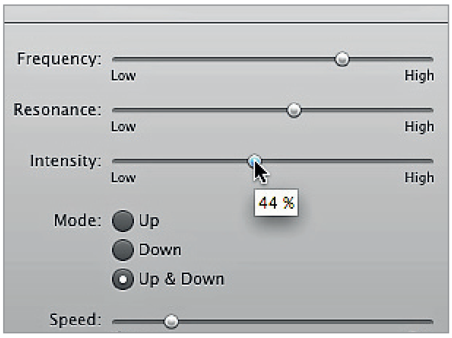



1 > Load the sample **Drums1.aif** from the **cm** DVD into your host and give it a listen to familiarise yourself with its unprocessed sound. It's a basic recording of a simple beat, with no treatment or effects. We're using Apple GarageBand for our example here, but you can use whatever host you like.

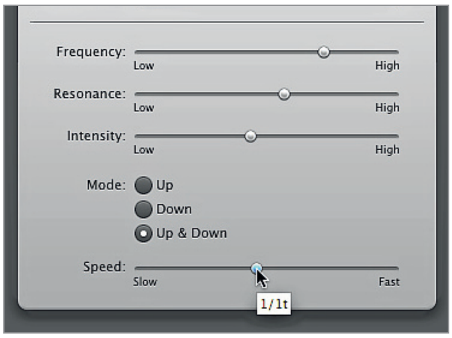

2 > As usual, you'll want to loop playback of the section you're working with, particularly since this is just a short loop. Insert a filter plug-in – it doesn't need to be anything too complicated. We're just using the Automatic Filter plug-in that comes with GarageBand.



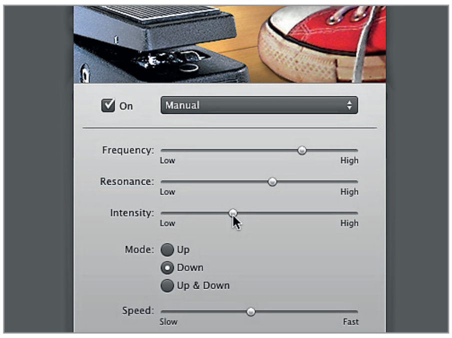

3 > Adjust the **Frequency** slider to hear the effect. Set it to around **70%**. You can clearly hear that everything above the cutoff frequency is filtered out, just like a shelving EQ. Next, try boosting the **Resonance** – not too high, though, or it'll begin to self-oscillate. **50%** or so will do. This emphasises the frequencies surrounding the cutoff. It's a classic effect!




4 > As we've mentioned a few times already, some effects have built-in Low Frequency Oscillators. In the case of this one, the LFO is what puts the 'automatic' in Automatic Filter. The **Intensity** slider governs how much modulation is applied to the Frequency. Push it up to about **45%**.




5 > You can plainly hear that the LFO is causing the filter's Frequency to rise and fall according to the LFOs wave shape. We can adjust the rate at which it oscillates with the **Speed** slider. This particular LFO is latched to the song's tempo and the value is represented accordingly. Give it a nudge to hear the effect in all its glory.




6 > Finally, let's change the LFO's **Mode**. By default it's set to **Up & Down**. Change it to **Down**. Can you hear the difference? If not, boost the **Intensity** slider a little more until you do. Now try the **Up** mode. Not every filter will have a built-in LFO, but those that do can add a lot of movement to your tracks.

# Recommended free plug-ins

Just a few of the many effects available for download online

## Reverb

### Christian Knufinke Software SIR1
Convolution reverb for PC with some tricks up its sleeve to minimise the usually high CPU consumption associated with such processing.
Web www.knufinke.de

### KarmaFX Reverb
Part of a donationware bundle from the brains behind KarmaFX Synth. It's Windows-only, unfortunately, but a brilliant little plug-in.
Web www.karmafx.net

### Kjaerhus Audio Classic Reverb
Simple, smooth and low on CPU, this is one of many must-haves in the PC-only Classic series.
Web www.kjaerhusaudio.com

### Togu Audio Line TAL-Reverb
Cross-platform, at last! TAL-Reverb is easy to use, offers pre-delay and filtering, and can be found at the following location:
Web kunz.corrupt.ch

## Dynamics

### Audio Damage Rough Rider
Beautiful cross-platform compressor combing a modern sound with a vintage approach. Bursting with character and a doddle to dial in.
Web www.audiodamage.com

### Camel Audio CamelCrusher
Another cross-platform compressor that's combined with a filter and distortion.
Web www.camelaudio.com

### Digitalfishphones Endorphin
This freeware dynamics plug-in has classic status. Choose between vintage and modern compression styles and add some saturation.
Web www.digitalfishphones.com

## Delay

### e-phonic RetroDelay
A retro sounding delay for Windows with all the trimmings.
Web www.e-phonic.com

### GSi WatKat
The Wem Copicat is a classic tape-based echobox and is emulated superbly in the WatKat. Mac and Windows versions available.
Web www.genuinesoundware.com

### Leslie Sanford Sanford Delay
This Windows plug-in delivers all the basics in a beautiful package. Up to one second of echo and ping-pong panning.
Web www.lesliesanford.com

## Distortion

### Studio Devil BVC
The classic British valve sound in a simple, free and cross-platform plug-in.
Web www.studiodevil.com

### Shuriken Berrtill
Not your normal distortion, as this one is designed to sound like circuit bent hardware.
Web www.shuriken.se

### Smartelectronix Cyanide
Cyanide, the graphical waveshaper from SE's Bram, has graced many a mangled track.
Web bram.smartelectronix.com

## Filter/EQ

### Analog Industries Filterizer
Excellent Mac/Windows VST multiband filter.
Web old.analogindustries.com/mt/software.html

### Blue Cat Audio Triple EQ
This three-band parametric EQ is a fine example of Blue Cat's considerable talents.
Web www.bluecataudio.com

### Ohm Force Frohmage
Utterly devastating filter from one of the premier effects developers.
Web www.ohmforce.com

### Voxengo Overtone GEQ
Want it quick and easy? Get hold of this superb seven-band graphic EQ.
Web www.voxengo.com

## Miscellaneous

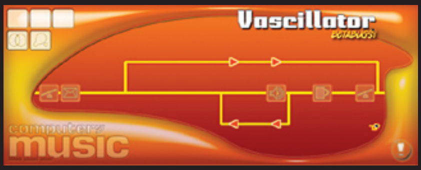
### Luxonix LFX-1310
Patterned after rack-mounted multi-effects hardware, this is practically a one-stop shop.
Web www.luxonix.com

### ndc Plugs Reversinator
It has one button and one light, and clicking that button does just what you think it does.
Web www.niallmoody.com/ndcplugs

### Smartelectronix KTGranulator
This plug-in from SE's Koen Tanghe will screw your stuff up in a big way. And that's a good thing. Granular processing for free!
Web koen.smartelectronix.com

### Valhalla DSP ValhallaFreqEcho
When we first discovered this free Bode-style frequency shifter, we could barely contain ourselves. Send your Cybermats into a tizzy.
Web www.valhalladsp.com


Apparently you *can* get something for nothing, as this free multi-effects box from Luxonix proves




Our very own and amazing Vascillator, as coded by the genii at Betabugs

## The cm Studio

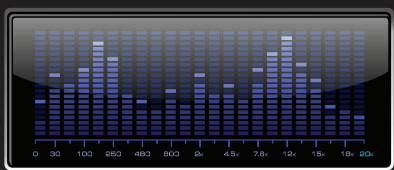
We've intimated throughout this feature that our very own cm Studio contains a plethora of useful, high-quality effects plug-ins, and each one is absolutely free with every issue of *Computer Music*. We're extremely proud of the collection – we've called upon some of the industry's best developers to bring you an utterly comprehensive range of processors that will serve the majority of your production and mixing needs.

So, what's included? Well, we've got an entire bundle of effects for guitarists in the shape of Guitar Suite CM. If that isn't enough grunge for you, you can dial up CMFuzz or the delightful CM WaveShaper. Pulse Modulator and Vascillator will get your toes tapping with some marvellous modulation, and you can tame your tracks with CompressiveCM. Create Cylon voices with our CM Vocoder and visualise your entire mix in an instant with FreqAnalyst CM. There's also the phenomenal Dynamic EQ, and a special edition of the awesome Sanford Phaser.

And then there's Ohmygod! What can we say about this twisted terror of a filter? Coming from Ohm Force, it's a modulatable comb filter that can be tweaked via MIDI or its onboard controls.

Yep, you get reverb, too. KR-Reverb CM Edition will put you in the perfect space, while Springverb calls up memories of that old broken Bi-Amp ambience generator. Finally, we couldn't bear to see you go without a decent delay, and KR-Delay cm Edition is a long way beyond merely 'decent'.

All of the above and much more appear exclusively on the cm DVD every month, and of course, the collection is added to on a pretty regular basis, so don't forget to check back on that **CM Studio** folder from time to time, even if you think you've already got everything in it.



# Effects tips for beginners

## DOUBLE YOUR PLEASURE!

**You can obtain a much thicker, more complex sounding reverb by chaining two reverbs in series. This is particularly effective when using two completely different reverbs, but you can always vary the parameters between two of the same model to obtain more interesting reflections. Simple reverbs can be made richer and more convincing using this technique, too. If you can't pile two chained 'verbs in your host's aux slots, try a plug-in chainer such as, er, Chainer from Xlutop.**

## DOUBLE DOUBLED!

**Just as combining two reverbs can add an extra layer of richness to your sound, so too can chaining multiple compressors. This can be a good way to achieve control over your signal's dynamics without bringing the unwanted artifacts of heavy compression. It's a great technique to try with lead vocals.**

## KNOW YOUR SPACE

**It's critical that you apply your reverb with discretion. The more reverb you put on something, the further away it will sound. While it might be OK to slather a big drum track with it, you should think twice about applying as much (or any) to the lead vocal if you want it to have some presence in the mix. This does *not* apply to shoegazing, My Bloody Valentine-type bands, however.**

## VOCODING

**Vocoders are, in actuality, complex, multiband envelope followers. A vocoder uses a modulation signal (like your voice) to shape a carrier signal (like your synth). Try using a vocoder to shape a choir sample for some Gregorian-style backing vocals!**

## PANORAMA

**Auto-panning is an effect that came into use in the 60s. Though it was absurdly overdone for a while, it seems to have been all but forgotten in modern production. Try using an auto-panner to throw some complex rhythms subtly around the stereo soundstage.**


**Synthi: This loopy instrument can be used as a filter and a delay, and there's even a built-in step sequencer**

## WE'VE GOT YOU SURROUNDED!

**Today's surround sound systems provide another exciting avenue for the creative mixologist. Many hosts support it and some plug-in effects are ideal for the job. Multi-tap echoes that bob around the listener's head; Phasers that sneak up from behind…**

## PSEUDO SUBS

**You're no doubt aware of pitch correction plug-ins, such as Auto-Tune, that recognise the pitch of incoming audio signals and shift them into perfect intonation. Some of these plug-ins have the ability to pump out MIDI notes and triggers that correspond to the pitch of the subject signal. Such a plug-in can be used to trigger a bass sound from your favourite synth, meaning that virtually any signal can be bolstered by the sub-oscillator of your choice.**

## THE ROUTE LESS TRAVELLED

**Creative effects routing can yield interesting results. For example, the conventional approach would dictate that delay and phasers comes before reverb, yet you can achieve some rich textural effects by putting the reverb first in a chain of effects.**

## MORE VOCODING

**Don't feel that you have to restrict your vocoder plug-in to the clichéd 'singing synth'. Vocoders can shape all sorts of sounds. For example, try uttering some wordless gibberish into your vocoder's modulator input and use it to modulate a found sound, such as crowd ambience or machine noises.**

## WE HAVE THE FORMULA

**If you have a favourite delay that doesn't sync to your host DAW's project tempo, figure out the desired delay rate by dividing 60,000 by the BPM. This will give you a value for quarter-note echoes in milliseconds. Divide that by two to get eighth-note divisions, and divide 40,000 by the BPM to get triplets.**

## STEPPIN' OUT

**We mentioned filters earlier, and you may have seen them with built-in LFOs or step sequencers that get the frequencies pumping. Why not try the same technique with other combinations – for example, use dedicated MIDI step sequencers and LFOs to automate any plug with MIDI inputs. cm**


**Send your music into a spin with this awesome auto-panner, Majken's PanOhRama!**